

*CONT
A4*

- 3 call to a plurality of pre-designated destination addresses which may be on the IP
- 4 network, on the PSTN, or on both the IP network and the PSTN.

- A5*
- 1 14. (Amended) A method of providing real-time voice communication between
 - 2 devices connected to an Internet Protocol (IP) network and devices connected to the
 - 3 public switched telephone network (PSTN), the steps of the method comprising:
 - 4 interfacing the digital data signals of the IP network with the voice signals
 - 5 of the PSTN;
 - 6 interfacing the control signals of the IP network with the PSTN to perform
 - 7 address translation, admission control, bandwidth management and zone management;
 - 8 routing calls between the devices connected to the IP network and devices
 - 9 connected to the PSTN;
 - 10 storing for each individual subscriber destination addresses on the PSTN
 - 11 and the IP network; and
 - 12 automatically routing calls to a subscriber to each destination address
 - 13 stored for that subscriber.

REMARKS

A request for a three month extension of time is submitted herewith. Claims 1-9 and 11-20 remain in this application. Applicant respectfully requests re-examination.

Claims 1-20 were rejected under 35 U.S.C. Section 102(e) as anticipated by *White, et al.* (6,069,890). Applicant respectfully traverses.

Claims 1 recites that the computer controlled switch is capable of receiving calls from the IP network or the PSTN and routing calls to the PSTN or IP network and "said computer controlled switch containing, for each subscriber, destination addresses on the

PSTN and the IP network; whereby calls to a subscriber received by the computer controlled switch are automatically routed to each destination address on the PSTN or the IP network for that subscriber". *White, et al.* fails to disclose such a capability. *White's* system describes a multi-services platform (MSP) 153 which will identify a specific subscriber's line and specify a ring count or interval for use in determining when a call to that subscriber has gone unanswered and should be forwarded to a voice mail system (Col. 14, Lines 1-6).

Method Claim 14 recites "storing for each individual subscriber destination addresses on the PSTN and the IP network; and automatically routing calls to a subscriber to each destination address stored for that subscriber". *White* does not teach or contemplate such a feature.

Furthermore, none of the references of record disclose or contemplate providing the apparatus of Claim 1 or the method of Claim 14.

The remaining claims in the application all depend from Claim 1 or Claim 14, respectively and add limitations thereto, making these claims patentable as well.

The Office Action questioned whether "PSTN" and "IP network" were juxtaposed in Claim 15. Applicant submits that the claim appears correct and requests that this objection be withdrawn.



Applicant respectfully submits that all the claims define over the prior art of record and are patentable over the prior art of record, and in light of the above amendment and remarks, respectfully requests that all the claims be allowed and this case passed to issue.

I hereby certify that this correspondence is being deposited with the United States Postal Service as First Class Mail in an envelope addressed to the Assistant Commissioner for Patents, Washington, D.C. 20231 on August 2, 2001.

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Respectfully submitted,

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VERSION WITH MARKINGS TO SHOW CHANGES MADE

IN THE SPECIFICATION

The last paragraph beginning on page 1 and continuing to page 2 has been amended as follows.

Digital transmission of information is much more desirable. Digital samples comprise one and zero bits. It is much easier to separate digital samples from line noise. Thus, when digital signals are regenerated, a clean sound can be maintained. As a result of the benefits of digital representation of the analog voice signals, pulse code modulation techniques were integrated into the telephone network. Pulse code modulation (PCM) converts analog sound into digital form by sampling the analog sound so many times per second and converting the sound into a numeric code. After the analog wave form is sampled, it is converted into a discrete digital form, as samples represented by code that indicates the amplitude of the wave form at the instant the sample was taken. A standard telephone form of PCM uses 8 bits for the code and a logarithm compression method that assigns more bits to lower amplitude signals. A standard transmission rate of 64K bits per second is used for one channel of telephone digital communication. The two basic variations of 64K bps PCM are [mu-log] μ -law and [a-log] A-law. Both methods are similar in that they both use logarithmic compression to achieve 12-13 bits of linear PCM quality with 8 bits. They differ in relatively minor compression details. North America uses [mu-log] μ -law modulation. Europe uses [a-log] A-law modulation. Another compression method that is often used today is an adaptive differential pulse-code modulation (ADPCM). A commonly used form of ADPCM is ITU-T G.726, which

encodes by using 4 bit samples giving a transmission rate of 32K bps. Unlike PCM, the 4 bits do not directly encode the amplitude of speech, but rather the differences in amplitude as well as the rate of change of that amplitude employing rudimentary linear prediction.

The second paragraph on page 3 has been amended as follows.

Another problem experienced in traditional toll networks is echo. Echo is normally caused by mismatch in impedance between the 4-wire network switch conversion to a 2-wire local loop. Although hearing your own voice in the receiver is common and reassuring to a speaker, hearing your own voice in a receiver longer [then 2.5] than 25 milliseconds will cause interruptions and breaks in the conversation. As a result, echo in the standard PSTN is controlled with echo cancelers and a tight control on impedance mismatches at the common reflection points. In packet based networks, echo cancelers are built into the low bit rate CODECs.

IN THE CLAIMS

The claims have been amended as follows.

- 1 1. (Amended) A system for providing real-time voice communication
- 2 between devices connected to an Internet Protocol (IP) network and devices connected to
- 3 a public switched telephone network (PSTN), comprising:
- 4 a computer controlled switch adapted for connection to a local public
- 5 switched telephone network and capable of receiving calls from the IP network or the
- 6 PSTN and routing calls to the PSTN or IP network; and
- 7 gate interface circuitry connected to the computer controlled switch and
- 8 adapted for connection to the IP network;

9 said computer controlled switch containing, for each subscriber,
10 destination addresses on the PSTN and the IP network;
11 whereby calls to a subscriber received by the computer controlled switch
12 are automatically routed to each destination address on the PSTN or the IP network for
13 that subscriber.

1 11. (Amended) The system of Claim [10] 1 wherein said computer controlled
2 switch receives an incoming call from the IP network or the PSTN and simultaneously
3 routes the call to a plurality of pre-designated [destinations] destination addresses which
4 may be on the IP network, on the PSTN, or on both the IP network and the PSTN.

1 14. (Amended) A method of providing real-time voice communication between
2 devices connected to an Internet Protocol (IP) network and devices connected to the
3 public switched telephone network (PSTN), the steps of the method comprising:

4 interfacing the digital data signals of the IP network with the voice signals
5 of the PSTN;

6 interfacing the control signals of the IP network with the PSTN to perform
7 address translation, admission control, bandwidth management and zone management;
8 [and]

9 routing calls between the devices connected to the IP network and devices
10 connected to the PSTN;

11 storing for each individual subscriber destination addresses on the PSTN
12 and the IP network; and

13 automatically routing calls to a subscriber to each destination address
14 stored for that subscriber.